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Hands-Fr ee Natural Language Processing System in Wireless Multimedia Personal Networks

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Abstract - The aim of this thesis work is to develop an automatic digital voice recognition program for English and convert to natural languages in wireless multimedia personal networks. The main purpose of this thesis work is to develop and implement Speaker Free Speech Recognition for sending Hands-Free and giving the user a Hands-Free application using GSM or Bluetooth technologies and system applications. Voice or speech recognition is the ability of a machine or program to receive and interpret commands or to understand and execute human speech commands. It allows us to communicate information effectively. Although online text and video resources greatly enhance the ability to communicate with one another through computer technology, speech is the first tool of communication used by humans for communication. This brief summary aims to set the context for subsequent conversations in one of the human language technologies: speech recognition (stated). Human languages are very different from machine languages. Speech recognition can also serve as a form of input. This can be very useful when one's hands or eyes are busy.

Index: Speech Recognition, Text Abbreviation, Mobile Devices, SR Database System, Natural Language Processing

1. Introduction

Speech recognition Automatic speech recognition or computer speech recognition converts spoken words into text. Like most desktop recognition software, the term voice recognition is sometimes used to refer to systems that require special speaker training. Identifying the speaker simplifies the task of translating the conversation. Speech recognition is a broader solution that refers to technology that can detect speech without targeting a speaker, i.e. a call center system that can detect arbitrary sounds. Speech recognition applications include voice user interfaces such as voice dialing, call routing, control of domestic devices, general data entry search, structured document preparation, and text processing from speech [1].

Speech Recognizer is a tool for identifying single spoken numbers. Another starting device is the IBM Shoebox, which recently demonstrated several improvements to a single system such as high system speed mass transcription capability such as the Sonic Extractor. One of the identifiable domains in the commercial application of speech recognition in the United States was the early sale of speech recognition (SR) to make the transcription process more efficient, in the opinion of health care, especially medical transcription (MT) work industry experts [2]. Hence not a complete transliteration. SRs at the time were often technically low. In addition, in order to use it effectively, physicians need to make changes in the way they work and record clinical encounters, many of which they are reluctant to do.

Speech recognition appears to be the biggest limitation of automated transcription software. The nature of the narrative command is very detailed, often requiring judgment given by a real man, but not yet provided by an automated system [3] . Another limitation is the extensive time required by the user and / or system provider to train the software. The difference in ASR between artificial syntax systems is that they are usually domain-specific and natural language processing, often language-specific. These types of applications offer their own specific goals and challenges.

2. Voice-tag technology

Voice-tag application is the process of converting human speech into abstract representation, and when users speak, this representation is used to identify and respond to the voice command that requires the same voice tag. Voice-tag applications use technologies such as Dynamic Time Warping (DTW) or HMM and are the first speech recognition to be introduced on embedded platforms such as mobile devices [4]. Users need to be able to speak words clearly on the microphone so that the app can clearly capture the sound and save it in the database to clarify the nature of their voice. An example is the voice dial feature of the iPhone that uses voice tag technology. Users should record their voice patterns for any contact in their address book, then dial the spoken voice pattern to record the number.

There are basically two types of speakers for voice tags / voice dial: speaker-based (recorded) and speaker-independent (recognizable). Speaker dependencies include voice dialing, name dialing and voice commands, which are a real feature on mobile devices today. For speaker dependents, users need to speak and record 1-3 times clearly to form voice tags, usually with a limited accent of a few words and limited vocabulary. (Representation of the grammar or syntax of the function) [5].

The phone responds to the recorded part, usually only when the same person who recorded them speaks. The advantages of speaker-based systems are high recognition accuracy, low complexity and easy access to voice feedback. Additionally, it is language free, so it is not limited to any particular language. Although speaker-independent, users do not need to be recording or training for voice tags, and the system does this automatically and transparently for users without any interaction [6].

For example, the user should talk directly to "everything" and then the cell phone will automatically find the most suitable for the spoken name from the phonebook of the user. The speaker-independent is language dependent and it increases the complexity. Thus, it is quite challenging because users should know and learn the utterance first, only then can language-specific methods be used to find out the pronunciation of the word. Large amounts of development are required for the pronunciation modelling for every supported language. Users who use the speaker independent voice recognition technique generally have a low recognition rate due to the different accents of users [7][8].

3. Hands-Free Natural Language Processing System

The Hands-Free Natural language messaging service is a text communication service component of phone, web or mobile communication systems that allows the transmission of short text messages between fixed line or mobile phone devices using fixed communication protocols. With 2.4 billion active users or 74% of mobile phone subscribers, Hands-Free is the most widely used data application in the world of text messaging. In many parts of the world, the term Hands-free is used synonymously with all types of short text messaging and user activity.

We have now implemented voice-based Hands-Free and PC control systems that are compatible with user voice input and voice databases that transfer user word acceptance from the device (MIC) to the message box. We also implemented a voice-based PC control system, which dictates the user's preferred system over voice

PC control. The system design consists of three modules, each of which is given in the following database management, voice HANDS-Free and voice based PC control.

This module acts as a voice input database, allowing only this type of grammar in the current dialect grammar database, for example the user King "King" application fits the system grammar and the output text is displayed as "right" or "rain" or some inconsistent word. Now, we propose to run a special database for Figure 1. ASR Speech recognition, this database module manages the words and names that the user normally uses. This module consists of three sub-modules, each of which lists the voice database management, user database and telecommunication settings.

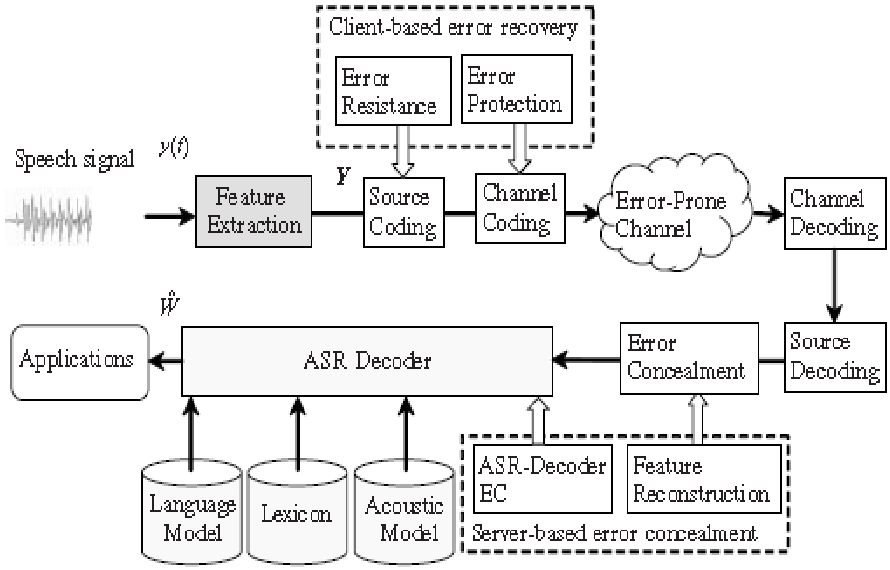

Figure.1: ASR Database System.

It is an important module for voice-based Hands-Free, converting user voice into text based on the application database, retrieving user voice input from the voice device (MIC), comparing the voice to the application database and matching it with the corresponding text message. This module also includes the Systematic Hands-free read option, where the user wants to input Hands-Free content, then activates the process and systematically reads and announces the Hands-Free content via some human voice. This module consists of three sub-modules, each of which lists an input voice message, an evaluation voice message, and a voice alert to assist the blind.

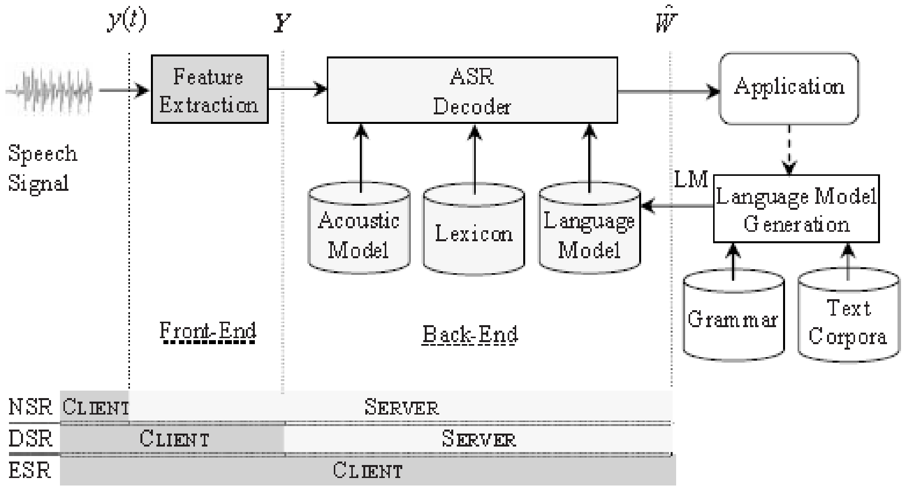

Figure 2 Architecture of DSR Voice System**.**

Voice Based Pc Control

It acts as a control PC application-based user voice command to manage MS Paint, MS Word and Notepad applications and to open the MR Polytechnics website based on voice input. Control input voice message, voice input and PC activity and applications. Although speaker-independent, users do not need to be recording or training for voice tags, and the system does this automatically and transparently for users without any interaction. For example, the user needs to speak "Anne" directly, and then the cell phone automatically finds a suitable match for the name spoken from the user's phonebook. The absence of a speaker depends on the language, which increases the complexity. Therefore, it is very challenging because users must first know and learn the pronunciation, and only then can they find the pronunciation of the word using specific methods of language. Each supported language requires a large amount of development for pronunciation modeling. Users who use the speaker free voice recognition technique generally have a lower recognition rate due to the different voices of the users.

4. **Part-Of-Speech Tagging Technology**

A ready-to-speak (POS) tagger that can tag a sentence or identify words and symbols in text with appropriate POS labels. Part of speech tagging is the process of determining the morph synthetic category of each word in a sentence and speaking a part of each word such as noun, verb, pronoun, preposition, adjective, adjective or other lexical class marker. A sentence. Tags play an important role in natural language applications such as data recovery, information extraction, speech recognition and natural language parsing. The string of words in the natural language sentence and the specific tag set become the input of the tagging algorithm, the output becomes the good POS tag. Each tagger has different language families so there is a standard tag set.

As a result, the tag set is a set of tags created by Tagger to choose from to attach to the corresponding word. There are good tag sets like N (noun), V (adverb), ADJ (adjective), ADV (adjective), PREP (preposition), CONJ (adjective) or NNOM (noun-noun). NSOC (name-social), VFIN (action limit), VNFIN (action non-finite), etc. Figure 4 shows an example of text tagging. The structure built by POS Tagore includes tokenization, ambiguity search and ambiguity solution.

Tokenization means dividing the text into tokens for further analysis, which can be pronounced as boundaries, words, and punctuation. An obscure search dictionary (list of word forms and parts of their speech) is also used by an essay dealer to search for unknown words. Finally, the opacity resolution (deviation) depends on the information about the word, contextual information, or word tag sequences. Tagging: Lesson sample taggers can be classified as rule-based tags and random taggers.

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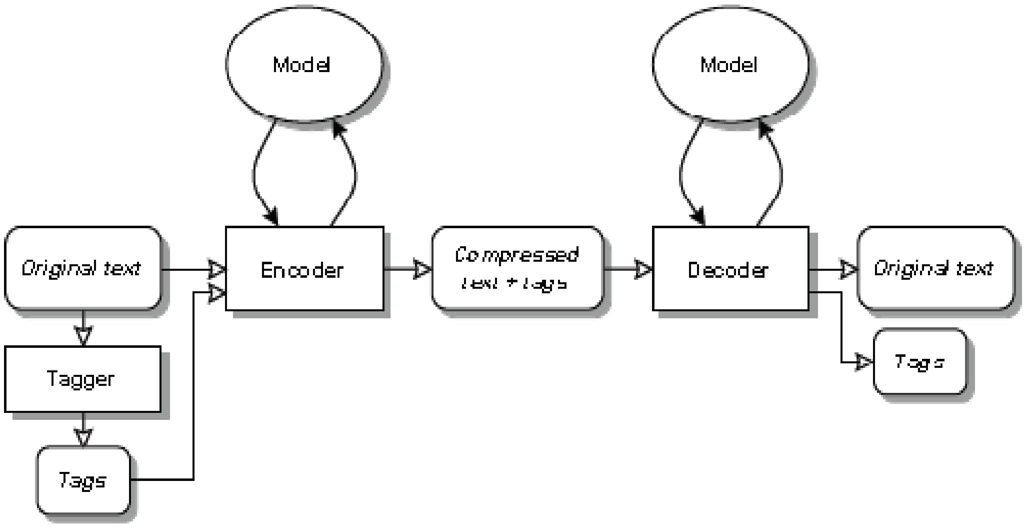

Figure 3 Model of tagging texts

Rule-based tagger is the oldest method of using a set of rules, usually to separate tag ambiguity. It usually gives possible tags for a word depending on the dictionary or dictionary. Tagged corpus, rules can be coded manually or retrieved from data. However, if a larger POS set is specified, the number of rules will increase significantly and the definition of the rule will become more expensive and difficult. The tag changes by examining the linguistic features of the word, its previous word, the following word and other elements. For example, if the previous word is an article, the word in question must be a noun.

Information about how Tagger works can be coded in the form of rules, which can be context-pattern rule or integrated into finite-state automata as general expressions that are literally associated with ambiguous sentence representations. Finally, the specific framework for iSay-Hands-free. In the end, we chose to implement ESR (embedded speech recognition) to iSay-Hands-Free because it is fully embedded in the target mobile. Voice-tags and Part of Speech (POS) tagging technology for sending short Hands-Free will be integrated into iSay-Hands-Free as a speech recognition solution.

For one thing, the concept of voice-text tagging for voice dial purposes is implemented on most cell phones today. Therefore, this concept has already proven to be functional, the same concept we used when implementing Hands-Free when implementing iSay-Hands-Free for users to save personalized text abbreviations. In short, it allows speech recognition to translate speech into text abbreviations that users prefer.

5. Conclusion

This is considered to be the most crucial step in getting a successful new system and giving it to the customer, giving them confidence that the new system will work and be effective. The implementation phase involves careful planning and investigation of the existing system, including barriers to implementation, design of methods to achieve change and evaluation of change methods. Implementation is the process of converting a new system design into a function. This step focuses on user training, site preparation and file conversion to install the candidate system. The important point to consider here is that the transition should not interfere with the performance of the organization. The app can make further improvements so that the app works in a much more attractive and useful way than it currently does. The speed of transactions is now more than enough. In the future, this project will cover almost all requirements. Since coding is primarily structured or modular in nature, further requirements and improvements can be easily made. Improvements can be made by replacing existing modules or adding new modules. An important development that could be added to this project in the future is the online voice-based hands-Free system using GSM technology, which is currently being used for offline voice-based Hands-Free systems using Bluetooth and mobile technologies.

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